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Network Coding Over SATCOM: Lessons Learned

Jason Cloud, Muriel Médard

Massachusetts Institute of Technology, Cambridge, MA 02139, USA,
jcloud@mit.edu, medard@mit.edu

Summary. Satellite networks provide unique challenges that can restrict users' quality of service. For example, high packet erasure rates and large latencies can cause significant disruptions to applications such as video streaming or voice-over-IP. Network coding is one promising technique that has been shown to help improve performance, especially in these environments. However, implementing any form of network code can be challenging. This paper will use an example of a generation-based network code and a sliding-window network code to help highlight the benefits and drawbacks of using one over the other. In-order packet delivery delay, as well as network efficiency, will be used as metrics to help differentiate between the two approaches. Furthermore, lessoned learned during the course of our research will be provided in an attempt to help the reader understand when and where network coding provides its benefits.

Key words: Intra-Session Network Coding, Implementation Concerns, Satellite Networks, In-Order Delivery Delay, Lessons Learned

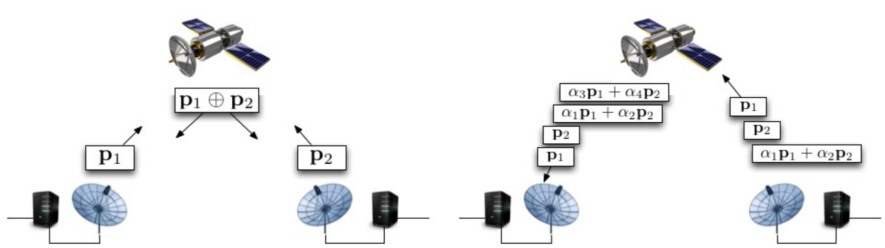
1.1 Introduction

Space-based packet data networks are becoming a necessity in everyday life, especially when considering world-wide Internet connectivity. It is estimated that over half of the world's population still does not have access to broadband Internet due to a variety of factors including a lack of infrastructure and low affordability, especially in rural areas and developing countries [1]. To overcome these barriers, a number of companies such as SpaceX, Google, and FaceBook have recently launched projects that incorporate some form of space-based or high altitude data packet network. However, significant challenges such as large latencies, high packet erasure rates, and legacy protocols (e.g., TCP) can seriously degrade performance and inhibit the user's quality of service. One promising approach to help in these challenged environments is network coding. This paper will investigate some of the gains that net-

work coding provides, as well as outline some of the lessons learned from our research.

Space-based networks have a number of unique characteristics that challenge high quality of service applications. Large packet latencies and relatively high packet erasure rates can negatively impact existing protocols. Fading due to scintillation or other atmospheric effects are more pronounced than in terrestrial networks. The high cost in terms of both deployment and bandwidth make efficient communication a requirement. Finally, the broadcast nature of satellite networks create unique challenges that are non-existent in terrestrial networks. While existing physical and data link layer techniques help improve performance in these conditions, we will show that coding above these layers can also provide performance gains.

Various forms of network coding can be used with great benefits in space-based networks. In general, these can be characterized into two broad categories: inter-session network coding, and intra-session network coding. Figure 1.1 provides a simple example of both. Inter-session network coding combines information flows together to improve the network capacity. A summary of the various methods that can be used for satellite communications is provided by Vieira *et al.* [2]. Intra-session network coding, on the other hand, is used to add redundancy into a single information flow. Adding this redundancy has shown that file transfer times can be decreased for both multicast [3] and unicast [4, 5] sessions.



(a) Inter-Session (b) Intra-Session

Fig. 1.1. Examples of inter-session (a) and intra-session (b) network coding. This paper focuses completely on intra-session network coding.

While there are merits to both techniques, our focus will be on intra-session network coding techniques that help achieve the following goals: provide consistent performance for protocols not designed for space systems; decrease delay for real-time or near real-time data streams; efficiently use any network resources that are available; and reduce packet erasure rates due to correlated losses. A generation-based approach [5, 6] and a sliding-window approach [7] will be used to help highlight the potential gains, design choices, and implementation decisions that need to be taken into account. Several performance

metrics including the in-order delivery delay, efficiency, and upper layer packet erasure rates will be used to help differentiate between the approaches.

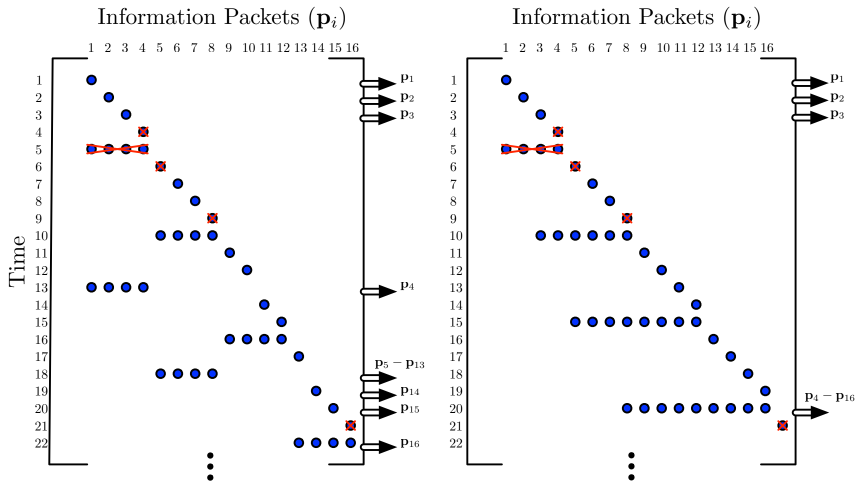
The remainder of the paper is organized as follows. Section 1.2 will provide details on the coding algorithms considered. Section 1.3 provides information about the assumed network model and evaluates the performance of these coding algorithms when used for both reliable and unreliable data streams. Section 1.4 discusses various considerations that need to be taken into account when implementing network coding into real systems. Finally, conclusions are summarized in Section 1.5.

1.2 Network Coding over Packet Streams

Network coding has been shown to dramatically improve network performance; however, implementing it can be a challenge. In order to develop practical coding techniques, random linear network coding (RLNC) [8] has been used by a large number of coding schemes because of its simplicity and effectiveness in most network scenarios. While both practical inter and intra-session techniques have been proposed, we are primarily interested in the latter due to the inherent limitations of existing satellite communication networks (i.e., typical satellite communication networks employ a bent-pipe architecture or have very limited on-orbit processing power). Assume that we want to send a file consisting of information packets $\mathbf{p}_i$, $i \in \mathcal{P}$, where $\mathcal{P}$ is the set of information packet indexes (i.e., the file has size $|\mathcal{P}|$ packets). Within these intra-session packet streams, RLNC can be used to add redundancy by treating each $\mathbf{p}_i$ as a vector in some finite field $\mathbb{F}_{2^q}$. Random coefficients $\alpha_{ij} \in \mathbb{F}_{2^q}$ are chosen, and linear combinations of the form $\mathbf{c}_i = \sum_{j \in \mathcal{P}} \alpha_{ij} \mathbf{p_j}$ are generated. These coded packets are then inserted at strategic locations to help overcome packet losses in lossy networks.

Management of the coding windows for these intra-session network coding schemes generally fall within the following two categories: fixed-length/generation-based schemes, or variable/sliding window based schemes. Fixed-length or generation-based schemes first partition information packets into blocks, or generations, $G_i = \left\{\mathbf{p}_{(i-1)k+1}, \ldots, \mathbf{p}_{\min(ik,|\mathcal{P}|)}\right\}$ for $i = [1, \lceil |\mathcal{P}|/k \rceil]$ and generation size $k \geq 1$. Coded packets are then produced based on the information packets contained within each individual generation. As a result, coded packets consisting of linear combinations of packets in generation G_i cannot be used to help decode generation $G_j, i \neq j$. Alternatively, sliding window schemes do not impose this restriction. Instead, information packets are dynamically included or excluded from linear combinations based on various performance requirements.

Examples of both schemes are provided in Figure 1.2. Columns within the figure represent information packets that need to be sent, rows represent the time when a specific packet is transmitted, and the elements of the matrix indicate the composition of the transmitted packet. For example, packet $\mathbf{p}_1$ is



(a) Generation-Based Code (b) Sliding-Window Code

Fig. 1.2. Examples of generation-based and sliding-window network coding schemes. It is important to note that the generation-based coding scheme requires feedback and retransmissions to ensure reliable delivery while the sliding-window coding scheme only requires feedback to help slide the coding window.

transmitted in time-slot 1, while coded packet $\mathbf{c}_5 = \sum_{i=1}^{4} \alpha_i \mathbf{p}_i$ is transmitted in time-slot 5. The double-arrows on the right of each matrix indicate when an information packet is delivered, in-order, to an upper-layer application, and the red crosses mark lost packets.

Each approach has its benefits and drawbacks. It is easy from a coding perspective to implement the generation-based coding scheme, and these schemes achieve capacity when $k \to \infty$. However, partitioning packets into generations adds artificial restrictions on the code's capability to recover from losses, and may not be as efficient as sliding window schemes. Furthermore, generation-based schemes can increase the complexity of the feedback process, especially for reliable data transfers. Sliding-window schemes, on the other hand, can outperform generation-based schemes in terms of efficiency and delay. Unfortunately, coding window management can be difficult and these schemes typically cannot guarantee a decoding event occurs before the termination of a session. In addition, the size of the coding window maybe much larger than the generation-based schemes leading to increased decoding complexity and communication overhead.

The examples shown in Figure 1.2 will be used throughout the remainder of this paper in order to provide some intuition into the trade-offs of using one type of coding approach over the other. Algorithm 1 describes the packet generation policy for the generation-based scheme shown in Figure 1.2(a), while Algorithm 2 describes the policy for the sliding-window coding scheme

shown in Figure 1.2(b). Each algorithm uses a systematic approach where information packets $\mathbf{p}_i, i \in \mathcal{P}$, are first sent uncoded and redundancy is added to help correct packet erasures by inserting coded packets into the packet stream. We will assume that the amount of redundancy added to the packet stream is defined by $R \geq 1$ (e.g., the code rate is $c = {}^1/_R$).

Algorithm 1: Generation-based coding algorithm [6]

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for each $j \in \left[1, \lceil \frac{|\mathcal{P}|}{k} \rceil\right]$ **do**
  $w_l \leftarrow (j-1)\,k + 1$
  $w_u \leftarrow \min(jk, |\mathcal{P}|)$
  for each $i \in [w_l, w_u]$ **do**
    Transmit $\boldsymbol{p}_i$
  for each $m \in [1, k\,(R-1)]$ **do**
    Transmit
    $\boldsymbol{c}_{j,m} = \sum_{i=w_l}^{w_u} \alpha_{i,j,m} \boldsymbol{p}_i$

Algorithm 2: Sliding window coding algorithm [7]

Initialize $k = 1$, $u = 1$, and
$n = \frac{R}{R-1}$
for each $k \in \mathcal{P}$ **do**
  if $u < n$ **then**
    Transmit packet $\boldsymbol{p}_k$
    $u \leftarrow u + 1$
  else
    Transmit $\boldsymbol{c}_k = \sum_{i=1}^{k} \alpha_{k,i} \boldsymbol{p}_i$
    $u \leftarrow 1$

It is important to note that feedback is not addressed in these algorithms. In general, feedback is necessary to accurately estimate the network packet erasure rate. Furthermore, feedback maybe required to ensure reliable delivery in some instances. For the generation-based scheme, the server may need to know the number of received degrees of freedom from each transmitted generation. This feedback can be used by the server to retransmit additional degrees of freedom if a particular generation cannot be decoded. Details are provided in [6]. In the sliding window scheme, knowledge of the number of received degrees of freedom may not be necessary [7]; but feedback can be used to help slide the coding window or facilitate decode events if there are delay constraints.

1.3 Network Coding Performance for Packet Streams

As we mentioned in the previous section, we will compare the performance of two types of intra-session network coding schemes (see Algorithms 1 and 2) for both a reliable data stream (e.g., a TCP session) and an unreliable data stream (e.g., a UDP session). The metrics used to evaluate both coding schemes will depend slightly on the type of data stream; however, the following definitions will be used throughout this section.

Definition 1. *The in-order delivery delay D is the difference between the time an information packet is first transmitted and the time that the same packet is delivered, in-order.*

Definition 2. *The efficiency η of a coding scheme is defined as the total number of degrees of freedom (i.e., the total number of information packets) that need to transfered divided by the actual number of packets (both uncoded and coded) received by the sink.*

Both of these metrics are particularly important for satellite communication systems. In the case of reliable data streams, large propagation delays can compound the effects of packet losses by creating considerable backlogs and in-order delivery delays. For large file transfers or non-time sensitive applications, this may not be an issue. However, a large number of time-sensitive applications (e.g., non-real-time video streaming) use TCP. Lost packets can result in very large resequencing delays that can seriously degrade the quality of user experience. Network coding is particularly useful in these situations to help recover from packet losses without excessive retransmissions. Furthermore, bandwidth is expensive for these systems. Any coding scheme that promises to provide a specified quality of service needs to be efficient.

The remainder of this section will provide an outline of the network model and examine the performance of the two coding schemes presented above. The two metrics defined earlier will be used in addition to any additional metrics that are important for the specific type of data stream.

1.3.1 Network Model

We will assume a time-slotted model where each time-slot has a duration t_s equal to the time it takes to transmit a single packet. The network propagation delays will be taken into account by defining $t_p = {}^{RTT}/_2$ where RTT is the round-trip time. As a reminder, we will assume that the amount of redundancy added ($R \geq {}^1/_{1-\epsilon}$ given that ϵ is the packet erasure probability) defines the code rate $c = {}^1/_R$. For the generation-based scheme, c is equal to the generation size divided by the number of degrees of freedom transmitted for that generation (i.e., $c = {}^k/_{Rk}$ where k is the generation size). In the case of the sliding window scheme, c is dependent on the number of consecutively transmitted information packets (i.e., $c = {}^{n-1}/_n$ where $n = {}^R/_{R-1}$ is the number of packets between each inserted coded packet).

The satellite channel will be modeled using a simple Gilbert channel with transition probability matrix

$$P = \begin{bmatrix} 1-\gamma & \gamma \\ \beta & 1-\beta \end{bmatrix} \tag{1.1}$$

where γ is the probability of transitioning from the "good" state (which has a packet erasure rate equal to zero) to the "bad" state (which has a packet erasure rate equal to one) and β is the probability of transitioning from the "bad" state to the "good" one. The steady-state distribution of the "bad" state $\pi_B = {}^\gamma/_{\gamma+\beta}$ and the expected number of packet erasures in a row $\mathbb{E}[L] = {}^1/_\beta$

will be used as the primary parameters for determining the transition probabilities of the channel model. It should be noted that this model does not necessarily reflect the effects of fading due to scintillation or rain, which generally have a duration equal to hundreds of milliseconds to hours. Instead, the model is intended to help model the cases where the SNR is such that the performance of the underlying physical layer code is degraded; but the situation does not warrant the need to change to a more robust modulation/coding scheme.

Lesson Learned: Network coding is not a cure-all solution. It cannot mitigate the effects of deep fades with very large durations.

1.3.2 Reliable Data Stream Performance

Reliable data delivery is a fundamental requirement for some applications. This section will focus on the performance of both a generation-based and a sliding-window coding scheme by looking at the following metrics: the ability of the scheme to provide 100% reliability, the in-order delivery delay, and the coding schemes' efficiency. Furthermore, the performance of an idealized version of selective-repeat ARQ will be provided to highlight the gains network coding can provide in satellite communications systems.

Before proceeding, feedback maybe necessary to ensure reliability. With regard to the two example coding schemes presented here, the generation-based scheme requires feedback while the sliding-window scheme does not. Algorithm 1 can be modified to include this feedback with only a few changes. Assume that delayed feedback contains information regarding the success or failure of a specific generation being decoded by the client. If a decoding failure occurs, the server can then produce and send additional coded packets from that generation to overcome the failure. On the other hand, the construction of the sliding-window scheme outlined in Algorithm 2 has been shown in [7] to provide a finite in-order delivery delay with probability one. Therefore, our results will assume that no feedback is available when using this scheme even though feedback may actually increase the algorithm's performance.

A detailed analysis of the in-order delivery delay and the efficiency for the generation-based scheme ($\mathbb{E}[D_G]$ and η_G respectively) is provided in [6], while the same is provided in [7] for the sliding-window scheme ($\mathbb{E}[D_S]$ and η_S respectively). The analysis of the generation-based scheme shows that $\mathbb{E}[D_G]$ and the delay's variance σ_G^2 are dependent on both the generation size k and the amount of added redundancy R. For a given R that is large enough and independent and identically distributed (i.i.d.) packet losses, $\mathbb{E}[D_G]$ is convex with respect to k and has a global minimum. Determining this minimum, $\mathbb{E}[D_G^*] = \arg\min_k \mathbb{E}[D_G]$, is difficult due to the lack of a closed form expression; however it can be found numerically. The following results will only show $\mathbb{E}[D_G^*]$ for a given R since the behavior of $\mathbb{E}[D_G]$ and σ_G^2 as a function of k is provided in [6]. The analysis of the sliding-window scheme's in-order delivery

delay shows that $\mathbb{E}[D_S]$ is only dependent on R since there is no concept of generation or block size. Therefore, a simple renewal process can be defined and a lower-bound for the expected in-order delay can be derived. While the efficiency of this scheme is not explicitly given in [7], it can easily be shown that the efficiency is $\eta_S = {}^1/_{R(1-\epsilon)}$ for i.i.d. packet losses that occur with probability ϵ. Regardless of this existing analysis, the in-order delivery delay and efficiency used below for both the generation-based and sliding-window coding schemes are found using simulations developed in Matlab.

Figures 1.3 and 1.4 show $\mathbb{E}[D]$ and η respectively for both coding schemes as a function of R. Furthermore, each sub-figure shows the impact correlated losses have on the schemes' performance where $\mathbb{E}[L]$ is the expected number of packet losses that occur in a row. For uncorrelated losses (e.g., $\mathbb{E}[L] = 1$), both coding schemes provide an in-order delivery delay that is superior to the idealized version of selective repeat ARQ. This performance gain becomes less pronounced as $\mathbb{E}[L]$ increases. In fact, the sliding-window coding scheme performs worse than ARQ for small R when $\mathbb{E}[L] = 8$. The cause of this is due to the lack of feedback, which can help overcome the large number of erasures if it is implemented correctly. Regardless, Figure 1.3 shows that coding can help in the cases where losses are correlated; although the gains come with a cost in terms of efficiency.

Lesson Learned: While feedback is necessary for estimating the channel/network state, it also aids in decreasing in-order delivery delay.

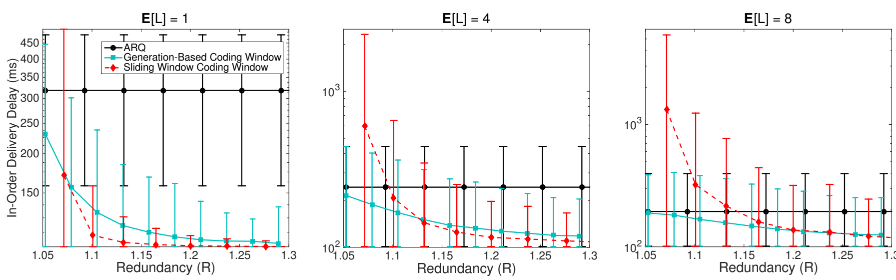

Fig. 1.3. In-order packet delay ($\mathbb{E}[D]$) as a function of the redundancy (R) where $RTT = 200$ ms, $t_s = 1.2$ ms, and $\pi_B = 0.05$.

Decreasing $\mathbb{E}[D]$ results in decreased η, which can be observed in Figure 1.4. The figure shows that the sliding-window coding scheme is more efficient than the generation-based scheme. There are two major contributors to this behavior. First, code construction has a major impact on efficiency. Since coding occurs over more information packets in the sliding-window scheme, coded packets can help recover from packet erasures that occur over a larger

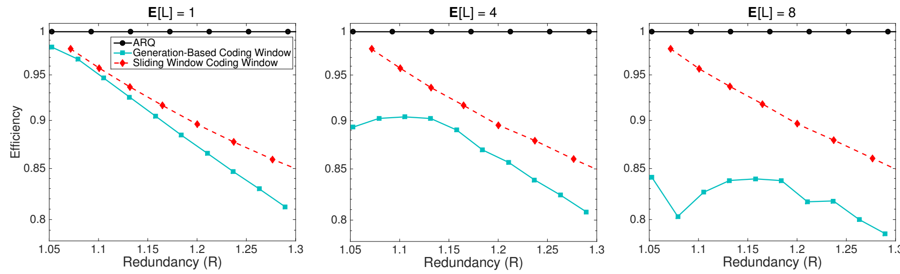

Fig. 1.4. Efficiency (η) as a function of the redundancy (R) where $RTT = 200$ ms, $t_s = 1.2$ ms, and $\pi_B = 0.05$.

span of time (i.e., multiple generations if we compare it with the generation-based scheme). Second, the decrease in the generation-based scheme's efficiency, as well as the non-decreasing behavior of η_G, for $\mathbb{E}[L] > 1$, is an indication that retransmissions are necessary to provide reliability. In fact, the generation-based scheme almost always requires retransmissions to be made when $\mathbb{E}[L] = 8$. This behavior helps illustrate that artificially restricting the coding window's size can have negative impacts and may not be the appropriate strategy in certain circumstances.

Lesson Learned: Generation-based coding schemes perform poorly when packet losses are correlated due to the limited number of packets that are used to form a coded packet.

1.3.3 Unreliable Data Stream Performance

Data streams such as real-time voice and video do not necessarily require 100% reliability. However, decreasing the underlying packet erasure rates may still drastically improve upper layer quality of service. Recent work in this area has shown that network coding is one tool that can help improve performance [9, 10]. This section will compare both the generation-based and sliding-window coding schemes with respect to the upper-layer packet erasure probabilities and the expected in-order delivery delays.

The generation-based coding scheme shown in Algorithm 1, where feedback is only necessary to identify the packet erasure rate, is ideally suited to the case where there is a delay constraint and packet delivery is not guaranteed. Packets within each generation are delivered in-order until the first packet loss is encountered. Once the entire generation has been received, the client attempts to decode it. If the generation cannot be decoded, only the successfully received information packets are delivered. If the generation can be decoded, every information packet contained in the generation is delivered in-order.

Modifying the sliding-window coding scheme shown in Algorithm 2 for unreliable data streams is somewhat difficult. If a delay constraint exists, the coding window cannot be arbitrary changed to accommodate these constraints. For example, assume that a lost information packet $\mathbf{p}_i$ is no longer necessary due to its delivery time exceeding some specified value. One approach would be to move the left side of the coding window to the right so that $\mathbf{p}_i$ is no longer used in the generation of future coded packets (i.e., $\mathbf{c}_j = \sum_{k=i+1}^{j} \alpha_{j,k}\mathbf{p}_k$). In order for these new coded packets to be useful, the decoder must discard any coded packet containing $\mathbf{p}_i$ that it has already received. Not only does this decrease the efficiency of the coding scheme, but it also potentially increases the delay for subsequent packets $\mathbf{p}_j, i < j$. As a result, we will assume that Algorithm 2 is left unchanged in this scenario.

Lesson Learned: Great care must be taken when modifying a sliding-window coding schemes' coding window when trying to meet a delay constraint. Not doing so properly can lead to decreased efficiency and increased in-order delivery delay for subsequent packets.

Figures 1.5 and 1.6 show the expected upper-layer packet erasure rate (PER) and expected in-order delivery delay $\mathbb{E}[D]$ respectively for both the generation-based (GB) and sliding-window (SW) coding schemes. Three values of the expected number of packet losses in a row $\mathbb{E}[L]$ and two levels of efficiency η (indicated by the values shown in parentheses) are provided. Due to the sliding-window coding scheme's construction, the PER and $\mathbb{E}[D_S]$ are constant with respect to k.

These figures illustrate some of the trade-offs that need to be taken into account when selecting the appropriate code. First, the larger the generation size in the generation-based scheme, the better the error performance. This is expected since you are essentially averaging losses over more packets. However, the cost is increased latency. Second, correlated losses can have a significant impact on the performance of the generation-based code. This is a result of partitioning information packets into generations, which places artificial constraints the ability of the code to recover from packet losses. The sliding-window scheme has no such constraints. On the other hand, the redundancy inserted into the packet stream must be enough to ensure that any delay constraints are satisfied. For example, Figure 1.6 shows that $\mathbb{E}[D_S]$ and σ_S can be very large if your goal is to be highly efficient (e.g., $\eta_S \approx 0.97$). In order to match the delay of the generation-based code, a significant amount of redundancy must be added to the packet stream.

Lesson Learned: Decreasing the efficiency of sliding-window coding schemes is necessary to outperform generation-based schemes in terms of in-order delivery delay.

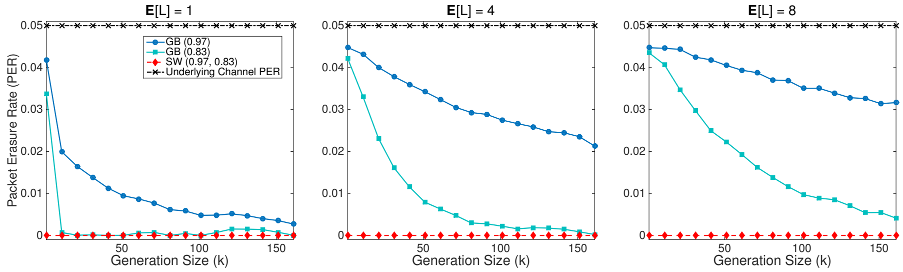

Fig. 1.5. Upper layer packet erasure rate (PER) as a function of the generation-based coding scheme's generation size (k) where $RTT = 200$ ms, $t_s = 1.2$ ms, and $\pi_B = 0.05$. The values shown within the parentheses for each item in the legend indicate the efficiency η.

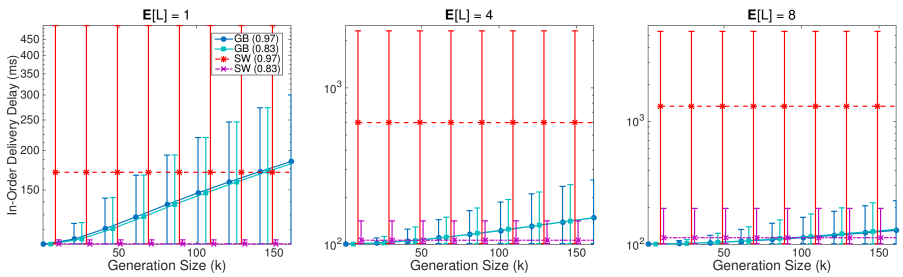

Fig. 1.6. In-order delivery delay $\mathbb{E}[D]$ as a function of the generation-based coding scheme's generation size (k) where $RTT = 200$ ms, $t_s = 1.2$ ms, and $\pi_B = 0.05$. The error bars show two standard deviations above and below the mean. The values shown within the parentheses for each item in the legend indicate the efficiency η.

1.4 Implementation Considerations

Implementing any type of network coding scheme presents its own challenges. Sections 1.2 and 1.3 highlighted just a few of them. However, there are a number of items that also affect how we code, especially in satellite networks. While we cannot address everything, we do provide a brief discussion on some of the items that we believe are important.

The first major consideration is where to perform the coding and decoding operations. Ideally, redundancy should be added at any point in the network where packet losses occur. This includes locations such as queues or links where the physical layer cannot provide 100% reliability. Furthermore, the amount of added redundancy should only be enough to help recover from losses that occur between network nodes that can code. This can be motivated by the simple example shown in Figure 1.7 where a source S wants to transmit $|\mathcal{P}|$ packets to the destination D. However, these packets must travel over a tandem network where each link $i \in \{1, 2, 3\}$ has an i.i.d. packet erasure probability ϵ_i. If end-to-end coding is used, $|\mathcal{P}| \left(\prod_i (1 - \epsilon_i)^{-1} - 1 \right)$ coded

packets must be generated at S and transmitted through the network. This results in an inefficient use of links closer to the source than would be necessary if redundancy is included into the packet stream at each node $R_i, i \in 1, 2$.

Lesson Learned: Coding at intermediate nodes, rather than coding end-to-end increases overall network efficiency.

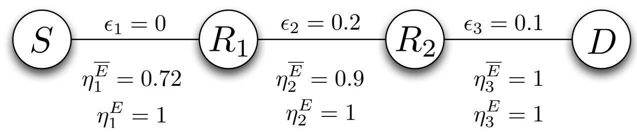

Fig. 1.7. A simple example showing that coding within the network is more efficient than end-to-end coding. η_i^j is the efficiency on link $i \in 1, 2, 3$ when coding is performed end-to-end ($j = \bar{E}$) or at each intermediate network node ($j = E$).

This simple fact can have major implications for satellite networks since bandwidth is limited and very expensive. As a result, coding should be performed at each satellite gateway or performance enhancing proxy (PEP) at a minimum; and if possible, at each hop in the satellite network. While coding should be performed as often as possible, network codes do not need to be decoded at each hop. This is also extremely beneficial in satellite networks since you can essentially shift a large portion of the required processing to the satellite gateway or end client. In other words, coded packets can be generated at multiple points within the network while only needing to decode once at the client or satellite network gateway. In the example provided in Figure 1.7, coding can take place at S, R_1, and R_2; however, only D needs to decode.

Lesson Learned: Decoding only needs to be performed once regardless of the number of times coding occurs within the network.

The second consideration that needs to be taken into account is how to communicate the coding coefficients α_i used to the decoder. For generation-based coding schemes where k is typically small, one can simply insert each coding coefficient into the header, which would require qk bits assuming each $\alpha_i \in \mathbb{F}_{2^q}$. Coding within the network only needs to modify the existing coefficients and does not increase the size of the coding coefficient vector. Of course, other approaches that require less than qk bits such as [11] or [12] can be used to decrease overhead.

Communicating the coefficients efficiently for sliding-window schemes is more challenging since the coding windows can be quite large. Existing meth-

ods typically use a pseudo-random number generator and communicate only the seed. This seed is then used by the decoder to generate the coefficients used to create each coded packet. Unfortunately, this does not scale well when coding occurs at intermediate network nodes. As an example, assume that an intermediate node's coding window contains multiple coded packets that were generated by previous nodes. When the node generates a new coded packet, it must communicate the seed used to generate the packet; in addition to all of the seeds for each of the coded packets contained within its coding window. If the coding window and the number of coded packets contained within the window are large, the amount of overhead required to reproduce the coefficients can far exceed the payload size.

Lesson Learned: The overhead required to communicate coding coefficents for sliding-window based schemes can be significant if not done correctly.

Finally, congestion control and file size can potentially dictate the coding approach used. Regardless of the type of data stream, some form of congestion control is typically needed at either the client/server or at the satellite network gateway. Common congestion control algorithms can cause bursts of packets, or packet trains, while they are ramping up to fully utilize the network. This behavior is even more pronounced when considering TCP flows over satellite networks. In these situations, it maybe preferable to use a coding scheme that provides a high probability of delivering every packet within a burst without needing retransmissions or waiting for the next packet burst to arrive. For example, a generation-based coding scheme can be used for small congestion window sizes and a sliding-window scheme can be used for large ones.

In a similar fashion, the coding strategy can also significantly impact the overall throughput for some file sizes. For example, consider a small file that can be transmitted using less than a single bandwidth-delay product worth of packets. A generation-based coding scheme, or a mixture of the generation-based and sliding-window schemes, should be used so that the the probability of decoding the file after the first transmission attempt is made very large. While this may impact the efficiency of the network, it can have major benefits for the user's quality of service or experience.

Lesson Learned: Congestion control and the length of the data stream may affect the network coding strategy.

1.5 Conclusion

Intra-session network coding is a promising technique that can help improve application layer performance in challenging space-based data packet networks. However, implementing it can be problematic if done incorrectly. This

paper used two common examples of intra-session network codes to show the benefits and drawbacks of one over the other. The first example used was a generation-based network code and the second a sliding-window based network code. While generation-based network codes are easier to implement, sliding-window network codes can provide improved performance in terms of in-order delivery delay and efficiency. This is especially the case when reliability is required. However, generation-based network codes are able to provide strict delay guarantees and improved upper layer packet erasure rates with little impact to the overall network efficiency when reliability is not a constraint. On the other hand, implementation considerations typically limit the performance of sliding-window network codes in these environments.

Lessons learned, as well as other implementation tips, were provided in addition to the above comparison. Some of the more important lessons learned include the facts that restricting the size of the coding window in any way limits the network code's performance gains; and feedback is useful for not only estimating the channel/network state information, but it also can be used to decrease delay. Both of these are apparent when considering the effects correlated packet losses have on the delay for reliable data streams. Various implementation considerations were also highlighted. These include where coding and decoding within the network should occur, how congestion control affects the way we code, and the challenges regarding the communication of RLNC coefficients between the source and sink. While properly implementing network coding in real networks can be difficult, we hope that our lessons learned will aid in the deployment of network codes in future satellite communication systems.

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